3rd GENERATION

Professional Audio Equipment

Audio Mixing – First Principles

Audio mixing of live and recorded music is an artform which, in its way, is every bit as creative as the musicians producing the original sound.

Just as a good musician needs the best instrument to project his creativity — so too does the sound engineer require superb quality audio equipment to enhance the musicians performance, and to mould the individual sounds into a composite whole. He needs to modify the instruments natural timbre via the equalisation stages to compensate for poor acoustics and route it if necessary to echo, flanging, chorus, A.D.T. or other effects units, and yet have total control of the dynamic range of the music via the faders to produce a balanced, and yet flexible stereo spectrum. Only good design, both ergonomically and electronically can fulfill all his requirements and allow him to concentrate solely on his task — whilst feeling totally at ease with the equipment he is using.

3RD GENERATION Professional Audio Equipment is designed to help the engineer to achieve his aims simply and quickly with the least possible fuss. It is manufactured using top quality components to be rugged, reliable and give years of trouble free service, and uses the latest technology to ensure that its performance will satisfy even the most critical ear.

In the following pages you will find operating instructions, comprehensive technical specifications, and block wiring diagrams for all of our products and systems together with a description of the basic principles of audio mixing and suggested stage layouts. We would ask you to remember that these are suggestions only. The permutations are virtually endless and a good sound engineers expertise together with a certain amount of experimentation will ensure that the best possible results are achieved from your 3RD GENERATION Professional Audio Equipment. And so read on . . .

Audio Mixers — Why Are They Necessary?

The function of an audio mixer is to collect sounds from various sources, e.g. musical instrument, vocalist, sound effect etc. and process it electronically both tonally and dynamically to give a total stereo sound balance. This balance may be natural or fed to external effects via the effect sends to modify the basic sound. The basic sounds are collected by two methods:

1. "Miking up" — this entails placing microphones in front of the vocalist, the individual on stage amplifiers, and around the drum kit.
2. Direct injection (abbreviated to "D.I.") — The sound is collected either prior to the on stage amplifier via a unit known as a "D.I. box", or from a signal taken from the amplifiers pre-amplifier stage. Once collected, each sound source is fed to a separate channel on the mixer. After being tone-corrected it is positioned in the stereo spectrum by way of the 'pan' control. When every channel has been adjusted the end result may be an exact reproduction of the stereo image on the stage, or the sound system could be used to completely change the apparent stage positions of the vocalists and instruments. It may even modify an instrument's sound or a voice via external effects units to bear no resemblance to the original. In the case of recording, the musicians would not be positioned in the studio as they would be on stage. In this instance the mixer could be used to synthesize the stereo spectrum so that it will appear to the listener that they are indeed hearing a 'live' recording.

The variations are potentially endless — this is the beauty of audio mixing. It may be used to naturally reproduce or enhance or modify the sound in countless ways which are limited only by the engineer's imagination and ability. Complete versatility is the keyword!

We will now proceed on to setting up and operating the sound system ...

The principles of setting up and operating of a sound system

The mixer is best situated approximately 25-40 metres from the stage, centrally placed in the auditorium for the correct stereo balance. Further back gives poor visual contact and a long delay of the sound between speakers and mixer. Try to avoid placing the mixer underneath or on a balcony as a poor representation of the actual sound leaving the speakers will reach the operator. Try and make all cables coming to and leaving the mixer as secure as possible, and if possible well above the audience.

All 3rd Generation ancillary equipment has earth — lift connectors on the back panels. To avoid earth — loop hum problems these may be lifted to isolate the internal electronic circuitry from the mains ground. Every piece of equipment will then be electronically earthed via the interconnecting screened cables, (Note — the mixer must be grounded to the mains earth at all times.)
All the equipment chassis can remain connected to the mains earth. This is obviously a much safer method of operation than relying on the screened cables to achieve a good mains earth.

Advised Microphone Techniques

Microphones should be low impedance, balanced types below 1000 ohms. You will find Cardioid characteristic microphones are the best to use for all P.A. work to avoid cross-talk and feed-back. Condenser microphones are useful for recording but tend to distort when used on stage because of the high sound pressure levels involved.

Lay out the microphones on stage to correspond with the mixer fader layout, i.e. left to right on stage and the mixer. This is especially important on vocal microphones to avoid confusion. Instrument amplification, electric pianos, synths, etc., may be miked but it is preferable to directly inject them into the mixer via a "D.I." box or straight from the pre-amplifier output which may be fitted to some amplifiers. Direct injected signals are always much cleaner because they avoid any distortion that may be introduced by using a microphone.
NEVER UNDER ANY CIRCUMSTANCES CONNECT AN AMPLIFIER'S LOUDSPEAKER OUTPUT TO THE MIXER'S INPUT.

Microphone Placement

The positioning of the microphones can be greatly influenced by variables — The natural acoustics of a concert hall, crosstalk between adjacent microphones, the Musicians personal sound preference etc. Therefore, the following advice should be taken as a very general guide only.
Bass drum mics should be placed about 200mm from the head. Damping of the heads to prevent ringing can be achieved by placing adhesive tape over sections of the head. If one head has been removed from the drum then a blanket or a cushion placed in the bottom of the drum and resting against the shell will have the same effect.
The snare drum mic should be placed about 50mm from the top head and near the edge to pick up some of the snare sound. Damping by adhesive tape may be needed.
Tom toms need to be miked about 25-30mm from the head and approximately 50mm in from the edge.
The hi-hat may need a mic placed about 50mm above the top cymbal.
Cymbals are best dealt with by placing a pair of mics above the entire kit. Known as overhead miking.
Synths, electric piano's and guitars can be directly injected or a mic may be placed about 200mm from any onstage amplification loudspeaker. The mic should be placed in the centre of the loudspeaker for the brightest sound or to the edge of the speaker for greater bass response.
Acoustic guitars can be miked close to the strings for a bright sound or near the sound hole for a mellow sound.
Vocal mics may sometimes need to be fitted with windshields.

Testing

Study thoroughly the connecting block diagrams. After you have interconnected the components of the system, and before it is turned on, *check the wiring thoroughly.* This is particularly important if you are using an electronic crossover, unit, as it is possible to accidentally connect the bass section to the horn drivers, which could result in permanent damage to them. It is good practice to disconnect the horn drivers prior to switching on the sound system as it is possible for the initial switch on 'thump' to damage them. Turn the equipment on in the following order. Mixer, graphic equalisers (if used), effects units, electronic crossover (if used) and finally the power amplifier. It is now safe to connect the horn drivers. To turn off, unplug the horn drivers first and then turn off the power amplifiers followed by the other equipment. DO NOT THROW ONE MASTER SWITCH WHICH SUPPLIES POWER TO THE WHOLE SYSTEM.

When the equipment is on, check that all amplifiers are functioning and check the loudspeakers and horns individually. If there appears to be a malfunction, you would be wise to suspect a fault in an interconnecting cable as most problems are relatively minor and cable faults are the main culprits. If this is not the case, you must isolate the fault. For instance, try connecting the speaker which is not working to a different amplifier.

If it works you should check the original amplifier with a loudspeaker and cable which you know to be working. The separate pieces of equipment can be eliminated in this way until you are left with one faulty item. Whilst it is not always possible to carry spares of everything, it is a good idea to carry a tool kit containing a mains test screwdriver, wire cutters, strippers, soldering iron and solder, pliers, spare plugs and connectors and most important of all, spare fuses. You may not be able to repair every possible fault but armed with the above you can remedy the minor ones (which most faults are), on the spot.

If everything is functioning you can now check the microphones to ensure that they are working and correspond with the layout on the mixer. At this point you should not be concerned with the sound but only that they are working. The microphone check is a job for two people. One person to talk into the microphones, the other operating the mixer.

Set all the faders to zero. Raise the input gain and use the P.F.L. monitoring system and headphones to listen to the microphones individually. When you are sure everything is working you are ready to commence the sound-check.

Soundcheck

This is the best way of testing the complete system. Your initial job is to obtain the main stereo mix. Ignore the foldback system and effects until this is achieved. First ask the musicians to play as they intend during the show, then with all faders still at zero, adjust the channel input gains by listening on the P.F.L. monitor and observing the peak indicators. The best signal to noise ratio is obtained by raising the input gain control until the peak indicator just begins to fire and then lowering it slightly. During the show, transients will probably cause the peak indicator to trigger occasionally but do not worry about this unless it is frequent and there is audible distortion as the indicator has been set to trigger slightly before an overload occurs. Up to this point the mixer has give you all the information you have needed — from now on it is mainly up to your ability and the quality of your hearing. Set all pan controls to mid-way, raise the master sliders, then raise the channel faders one at a time and adjust the tone controls. It will probably take some time to achieve a good sound and you may well find that you have to adjust each channel several times before you are satisfied. Keep an eye on the peak indicators as altering the tone controls may well necessitate a change on the input gain control. This may not necessarily mean a reduction!

Large systems use a stereo electronic crossover on the mixer outputs to split the sound into two or more sections (i.e. bass, middle, treble). Each individual sound is sent to an amplifier and loudspeaker cabinet which is specially designed to handle these particular frequencies. The crossover has a separate gain control on each section. These controls may be used to compensate for poor acoustics and to obtain a fairly flat response before you adjust the mixers tone controls. A certain amount of audio juggling will probably be required but the final result will be markedly superior to systems not using a crossover. Electronic crossovers should not be confused with the passive type of crossover found inside loudspeaker cabinets. Although their basic function is the same their method of operation is totally different and the electronic crossover is far more versatile than it's passive cousin. Once you are satisfied that everything is as it should be you can begin to position the channels in the stereo spectrum by means of the pan controls. One feature which is standard on the larger mixers and is a great aid to live mixing is the ability to route the input channels to sub-groups. For instance, channels 1-5 to sub-groups 1 and 2, channels 6-10 to sub-groups 3 and 4, and channels 11-16 direct to left and right outputs. What in effect this means, is that any channels which are routed, for example to sub-groups 1 and 2 are controlled by the faders of those sub-groups. If seven microphones had been placed around a drum kit and had been balanced by the seven channel faders, panned left and right, and routed to sub-groups 1 and 2, the overall kit level would be controlled by the faders sub-groups 1 and 2. Therefore if it became apparent during the show that the kit was not loud enough

it could be adjusted via the sub-group faders rather than the engineer trying to raise seven channel faders simultaneously whilst trying to maintain the ratios between them that had originally been set. The sub-groups may also be panned between the left and right outputs and obviously can be used to create some quite startling stereo effects.

For recording studio purposes there is a mono direct output from each sub-group which may be used in conjunction with a multi-track tape deck.

Now it only remains to add any special effects you require. The effects units are connected between the aux outputs and inputs. There are three aux sends on each channel of some of our mixers. One is pre-fader, one is switchable to pre or post fader and one is post-fader. On the other mixers there are four aux sends, two of which are switchable. It is usual to use the post-fader sends for effects and if more than one effect is needed then you may use one of the switchable sends set at the post-fader position. The master aux sends and returns may be monitored by the pre-fade listen. You should aim to set the send levels as high as possible without overloading the effects unit input and the returns as low as possible to achieve the best signal to noise ratio. Minor changes to effects may be made during the performance by the channel send levels. As the effects are being sent post-fader they will be altered in ratio to the main signal by changes to the channel fader level. If you need extra tone control over the effects and have a spare channel you may plug the effects return into this channel, set the gain by the method used for the microphone and utilise the channel's tone controls. In these circumstances you *must not* have this channel routed out through the effects send as you would create a loop which could cause the effects to feed-back on itself. Once you have reached this point and are satisfied with the results, your main problems are over. You should now turn to setting up the foldback. Decide how many foldback sends you need. If only one use the pre-fader aux send. If more than one you must use the pre-fader aux send and set the switchable send(s) to pre-fader. Not all channels may need to be routed back to the stage, in which case the relevant aux send channel levels should be set to zero. In other cases a channel may need to be routed to one foldback system and not theother(s), and so the aux level should be set accordingly. It is impossible to hear the onstage monitor mix from the mixer. Therefore you must use headphones and the aux send(s) pre-fade listen facility. Keep the master aux send levels low until you have achieved a good foldback balance and then raise them until feedback in near. Feed-back is a problem with foldback systems because of the proximity of the onstage monitor cabinets to the microphones. Many large sound systems utilise a graphic equaliser on the foldback sends which because of the very narrow operating band of its tone controls can 'notch out' feed-back without overly affecting the overall sound. The tone of the foldback is dictated by the tone settings on the mixer. These settings may sound excellent from the main P.A. loudspeakers but may not suit the foldback system. A graphic equaliser will enable you to adjust the foldback tone independently of that set on the mixer.

You are now ready for the show and may shut the system down as previously described until it is needed.

Pre-show check

Repeat the turning on procedure as in 'testing' and check the microphones. This can be done via the pre-fade listen facility and headphones. Listen to each microphone in turn. The ambient noise in the auditorium is usually enough to test the system. Allow plenty of time for this in case a fault has occurred and some last minute repairs are necessary.

Operation

At all times watch the peak indicators and the V.U. meters for any peaks which may be present. Any strange noises can be detected on the P.F.L. system. Make a point of monitoring the foldback system from time to time.

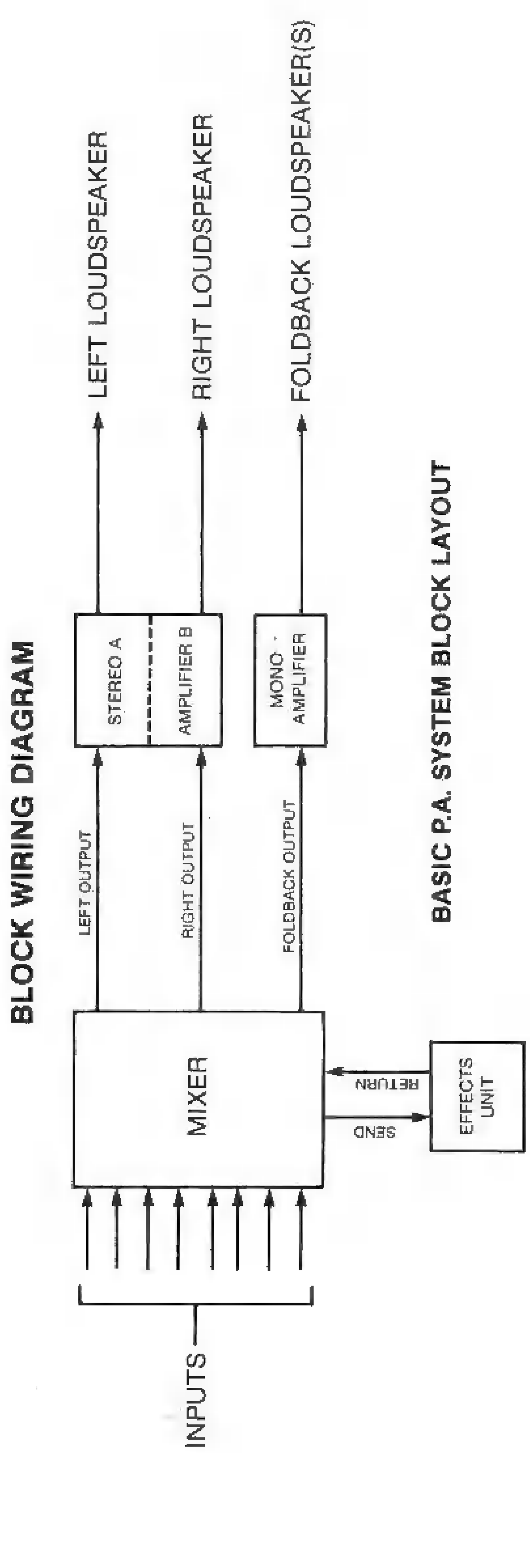
Do not be tempted to make too much use of stereo panning as the audience on one side of the auditorium may not be able to hear what is coming from the loudspeakers on the other side. For this reason voices and instruments should not be panned to the extreme left or right except for special effects. Moderation is the order of the day. Keep the main voice central and also the snare drum, bass drum and bass as these are the foundation upon which the music is built.

Many musicians improvise during a show and may take solos unexpectedly. You will probably have to raise the level of their instrument in the mix to highlight it. Always be prepared for the unexpected and observe the musicians closely.

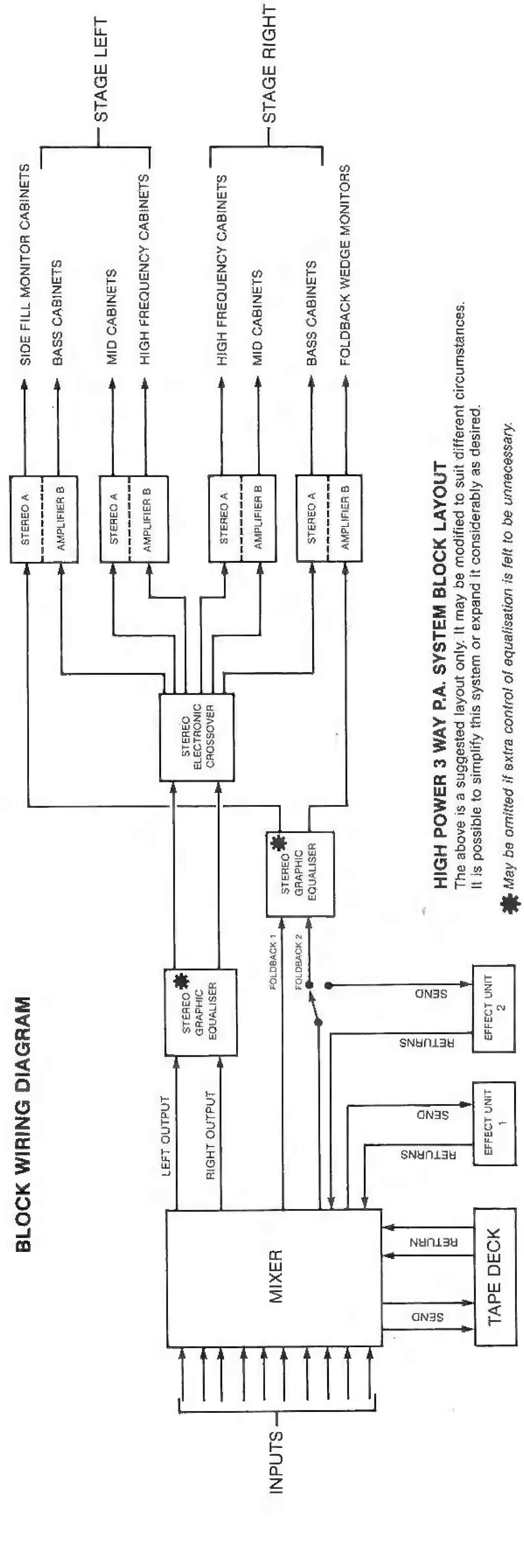
These instructions are intended as a guide only. A good sound engineer will experiment to find techniques of his own. He will not adhere rigidly to this advice but rely on his experience and ability to obtain the best results.

A sound system is an extremely versatile tool which in the hands of a craftsman can create an audio masterpiece. We wish you every success and are sure you will find our Professional Sound Equipment a pleasure to work with.

Sound System component interconnecting diagrams



BASIC P.A. SYSTEM BLOCK LAYOUT



HIGH POWER 3 WAY P.A. SYSTEM BLOCK LAYOUT
The above is a suggested layout only. It may be modified to suit different circumstances.
It is possible to simplify this system or expand it considerably as desired.

✱ *May be omitted if extra control of equalisation is felt to be unnecessary.*

Glossary of Terms

The following definitions refer to audio sound systems. Some are slang terms, some technical. They are not necessarily true grammatical definitions but are meant to help you to understand the operation of 3RD Generation Professional audio products and associated equipment.

AMBIENCE. The natural acoustics and resonance of an auditorium.
AUXILIARY. An alternative or secondary e.g. auxiliary output.
BALANCED LINE INPUT. A type of microphone input which has two 'hot' or 'live' connections and an earth. Any noise picked up by the microphone cable is automatically cancelled out.
BASS BIN. A type of loudspeaker cabinet designed to handle low bass frequencies only.
BULLET RADIATOR. A type of high frequency loudspeaker, the visible centre section of which is in the shape of the nose of a bullet.
CANS. Headphones.
CONDENSER MICROPHONE. A type of microphone which requires a voltage to be applied to it's 'hot' connections to polarize it. Usually very good quality but very easy to overload. See phantom power.
CROSSOVER. Used to split audio frequencies into bands, e.g. bass, middle, treble. There are two types. Active (or electronic) crossovers and passive crossovers. Their method of operation is totally different.
CROSS STAGE MONITOR. Loudspeaker cabinets placed at each side of the stage angled across it enabling musicians to hear each other.
d.B. A measurement of volume. Abbreviation of decibel.
D.I. Abbreviation for direct injection. A signal is fed straight into the mixer input from, for example, the output of an electric piano, synthesiser or guitar.
D.I. BOX. Splits the output from an instrument into two. One connects directly to the mixer input, the other connects to the onstage amplifier.
DIRECT INJECTION. See D.I.
DISTORTION. Caused when a signal level is too high and overloads an input or when a fault develops. It is recognisable as a harsh breaking up of the sound.
DYNAMIC MICROPHONE. A type of microphone which utilises a moving diaphragm to pick up sound.
E.Q. Abbreviation of equalization. A tone correction network.
EQUALIZATION. See E.Q.
FADER. A volume control which is a slide control as opposed to a rotary control.
FOLDBACK. On stage monitoring of the sound. The sound picked up on the stage and fed to the mixer is literally 'folded back' onto the stage again to enable the musicians to hear one another.
GRAPHIC EQUALIZER. A multi-section tone control using sliders instead of rotary controls. The positions of the slider knobs when viewed form a graph of the frequency response. Usually used on foldback systems to 'notch out' frequencies which are causing feedback.
HORN DRIVER. A type of loudspeaker which uses a very small diaphragm to reproduce sound (upper middle and high frequencies) and relies on a horn flare at it's front to increase it's efficiency.
LENS. Horn drivers and their associated flares are efficient but very directional. A lens is used at the front of the horn flare to disperse the sound. It usually consists of a series of horizontal metal plates angled downwards.
LINE INPUT. An input by which tapedecks, which usually have quite high signal levels, are connected to the mixer.
LINE OUTPUT. The output by which a tapedeck's input is connected to a mixer. This could be the sub-group outputs or the master left and right outputs.
MASTER. Has overall control, e.g. master fader.
MID BIN. A loudspeaker cabinet especially designed to reproduce middle frequencies.
MONITOR (To, Verb). To listen to e.g. headphone monitor — used to listen via the headphones to various sound sources on the mixer.
MONITOR (Noun). A loudspeaker cabinet used in foldback systems. See Wedge Monitor.
MULTI-CORE CABLE. A many stranded cable used to carry all the outputs and inputs between the stage and the mixer which is usually situated near the rear of the auditorium.
OVERLOAD. When a unit is asked to carry a signal of higher level than it can handle, then it is overloaded. This causes distortion.
PAN. An abbreviation of panoramic routing. Usually a continuously variable rotary control which swings a signal between the left and right of the stereo spectrum.
PARAMETRIC E.Q. A control which moves the operating band of a tone control up or down in the audio frequency range.
PEAK. The highest dynamic point of a sound.
P.F.L. Abbreviation for pre-fader listen. A point prior to the fader which may be listened to via the headphones. The fader will not affect the signal reaching the headphones.
P.F.M. Abbreviation for pre-fade monitor. Similar to P.F.L.
PHANTOM POWER. A facility which allows you to power condenser microphones directly from a power supply built into the mixer.
PRE-FADER. Access to a signal path before the fader.
POST-FADER. Access to a signal path after the fader.
RACK. A heavy duty transit case designed to hold and protect 19" rack mounted equipment e.g. power amplifiers, graphic equalizers and electronic crossovers etc.
SIDE FILL MONITORS. Similar to cross stage monitors.
SLIDER. Similar to a fader.
SNAKE. Similar to a multi-core cable.
STAGE-BOX. Connected to the stage end of a Snake fitted with connectors to carry inputs and outputs to and from the mixer to the stage.
SUB-GROUP. Situated prior to the master faders. Mixer input channels may be grouped together and assigned to sub-groups. Each sub-group may have overall control of a number of channels, e.g. seven microphones around a drum kit could be controlled by one sub-group fader.
SWEEP. See parametric.
TALK BACK. A microphone connected to the mixers talk back facility may be used by the engineer to communicate with the musicians or road crew on stage via the foldback loudspeaker cabinets.
WEDGE MONITOR. A loudspeaker cabinet positioned at the front of the stage by the musicians' feet for foldback monitoring. It is wedge shaped to enable the loudspeaker to be angled towards the listener.

3rd Generation Electronics are engaged in a continuous research and development programme and therefore reserve the right to change specifications without notice.

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